The embodiments of the invention in which an exclusive property or privilege is claimed are defined as follows:

1. An apparatus for controlling a gain of an audio signal broadcast by a loudspeaker into a room having ambient noise, comprising:

a first input for receiving the audio signal;

a second input for receiving an ambient room signal detected by a microphone, the ambient room signal having a noise portion representing the ambient noise and an audio portion representing the audio signal;

an adaptive filter for adaptively filtering the audio signal to generate a filtered signal that approximates the audio portion of the ambient room signal, the adaptive filter having a plurality of adaptation coefficients that are modified according to a predetermined formula having a threshold for limiting the rate of variation of the adaptation coefficients;

an error signal generator for subtracting the filtered signal from the ambient room signal to generate an error signal; and

a compander for adjusting the gain of the audio signal responsive of the error signal to compensate for the ambient noise.

2. The apparatus of Claim 1, wherein the adaptation coefficients are updated when the error signal level is above a predetermined fraction of the audio signal level.

3. The apparatus of Claim 1, wherein the adaptive filter is a variation of a normalized least mean square adaptive filter.

4. The apparatus of Claim 1, further comprising a down-sampler for sampling the audio signal at a sample rate lower than the highest frequency component of the audio signal, thereby generating a down-sampled audio signal, and wherein said adaptive filter adaptively filters the down-sampled audio signal to generate the filtered signal.

5. The apparatus of Claim 4, wherein the adaptation coefficients are updated according to the formula

$$Wi(n+1) = Wi(n) + \frac{\min[beta, e(n)] * u(n-i)}{epsilon * E(n)},$$

where $W_i(n)$ is representative of the i-th adaptation coefficient for the n-th audio sample, *e(n)* is representative of the error signal, *u(n-i)* is representative of the (n-i)th sample of the down-sampled audio signal, *E(n)* is representative of the energy level of the audio signal, *beta* is the threshold value, and *epsilon* is a stability coefficient.

6. The apparatus of Claim 5, wherein *E(n)* is calculated by generating a series of root-mean-square values of audio signal samples, and then filtering the series of root-mean-square values with a window function to generate a long term root-mean-square average to reduce the effect of sudden amplitude increase in the audio signal.

7. The apparatus of Claim 1, wherein the compander determines a target gain level *TG* for the gain of the audio signal according to the formula

$$TG = MinG + (GR * \min[1.0, (e - NT) / NR]),$$

where *MinG* is representative of a minimum gain level, *GR* is representative of a gain range, *e* is representative of the error signal level, *NT* is representative of a noise threshold, and *NR* is representative of a noise range level.

8. The apparatus of Claim 7, wherein the minimum gain level is approximately between –40 dB and 0 dB.

9. The apparatus of Claim 7, wherein the gain range is approximately between 0 dB and 40 dB.

10. The apparatus of Claim 7, wherein the noise threshold is approximately between –80 dB and 0 dB.

11. The apparatus of Claim 7, wherein the noise range is approximately between 1 dB and 60 dB.

12. The apparatus of Claim 7, wherein the compander adjusts the current gain applied to the music signal by increasing the current gain if the target gain is greater than the current gain.

13. The apparatus of Claim 12, wherein the rate of increase in the current gain is determined according to the formula

current gain = current gain + a1 / (Attack Time),

where a1 is a predetermined constant, and Attack Time represents the amount of time for the current gain to increase by 40 dB.

14. The apparatus of Claim 7, wherein the compander adjusts the current gain applied to the music signal by decreasing the current gain if the target gain is less than the current gain.

15. The apparatus of Claim 14, wherein the rate of decrease in the current gain is responsive to a user-defined parameter Release Time and determined according to the formula

current gain = current gain – a2 / (Release Time),

where a2 is a predetermined constant, and Release Time represents the amount of time for the current gain to decrease by 40 dB.

16. The apparatus of Claim 1, further comprising a non-volatile memory device for storing parameters used by the adaptive filter.

17. The apparatus of Claim 16, wherein the adaptive coefficients are initially determined according to a calibration procedure having an *Override* state that saves a set of parameter values currently used by the compander and in replacement thereof a set of default parameter values optimized for fast adaptation, a *Wait* state that adjusts and monitors the adaptation coefficients until the adaptation coefficients are optimized, a *Compute* state that computes a calibration value for use in preventing gain runaway, a *Save* state that saves the optimized adaptation coefficients in the non-volatile memory device, and a *Restore* state that restores the set of parameters saved during the *Override* state.

18. The apparatus of Claim 1, wherein the compander adjusts the gain of the audio signal when the error signal is above a predetermined noise threshold level.

19. The apparatus of Claim 18, wherein the predetermined noise threshold level is defined by a user.

20. The apparatus of Claim 19, wherein the user-defined noise threshold is increased to a Noise Threshold Override value when the user-defined noise threshold is lower than the Noise Threshold Override value, wherein the Noise Threshold Override value is computed by:

$$\text{Noise Threshold Override} = \max\left[\left(GR / \max(CR, 1.0) + MaxG + \max(CG, CI) + CAL\right), NT\right]$$

where GR is representative of a gain range, CR is representative of a compression ratio between the gain range and a noise range, MaxG is representative of a maximum gain, CG is representative of a compander gate, CI is representative of a compander input level, CAL is representative of a calibration value, and NT is the user-defined noise threshold.

21. The apparatus of Claim 20, wherein the calibration value *CAL* is computed by adding the RMS value of the error signal and the RMS value of a headroom signal, then minus the RMS value of the audio signal.

22. The apparatus of Claim 21, wherein the RMS value of the headroom signal is calculated by taking the difference between the RMS values of the ambient room signal and the error signal, multiply the different with a first constant and then minus a second constant.

23. The apparatus of Claim 22, wherein the first constant is approximately 0.6, and the second constant is approximately 3.0.

24. A user interface for calibrating an automatic volume control system that adjusts a gain of an input signal according to an ambient noise level, comprising:

a first control for adjusting a minimum gain level;

a second control for adjusting a gain range level;

a third control for adjusting a noise threshold level; and

a fourth control for adjusting a noise range level;

wherein the minimum gain level, the gain range level, the noise threshold level, and the noise range level are used by a gain computation procedure for determining the gain of the input signal.

25. The user interface of Claim 24, further comprising a first control for adjusting an attack time, and a second control for adjusting a release time, wherein the attack time and the release time are used by the gain computation procedure in determining the gain of the input signal.

26. A method for controlling a gain of an audio signal broadcast by a loudspeaker into a room having ambient noise, comprising:

receiving the audio signal;

receiving an ambient room signal detected by a microphone, the ambient room signal having a noise portion representing the ambient noise and an audio portion representing the audio signal;

adaptively filtering the audio signal to generate a filtered signal that approximates the audio portion of the ambient room signal, the adaptive filter using a plurality of adaptation coefficients that are modified according to a predetermined formula having a threshold for limiting the rate of variation of the adaptation coefficients;

using an error signal generator for subtracting the filtered signal from the ambient room signal to generate an error signal; and

using a compander for adjusting the gain of the audio signal responsive of the error signal to compensate for the ambient noise.

27. The method of Claim 26, further comprising updating the adaptation coefficients when the error signal level is above a predetermined fraction of the audio signal level.

28. The method of Claim 26, wherein the adaptive filter utilizes a variation of a normalized least mean square adaptive filter.

29. The method of Claim 26, further comprising down-sampling the audio signal at a sample rate lower than the highest frequency component of the audio signal, thereby generating a down-sampled audio signal, and adaptively filtering the down-sampled audio signal to generate the filtered signal.

30. The method of Claim 29, further comprising updating the adaptation coefficients according to the formula

$$Wi(n+1) = Wi(n) + \frac{\min[beta, e(n)] * u(n-i)}{epsilon * E(n)},$$

where $W_i(n)$ is representative of the i-th adaptation coefficient for the n-th audio sample, $e(n)$ is representative of the error signal, $u(n-i)$ is representative of the (n-i)th sample of the down-sampled audio signal, $E(n)$ is representative of the energy level of the audio signal, *beta* is the threshold value, and *epsilon* is a stability coefficient.

31. The method of Claim 30, further comprising calculating $E(n)$ by generating a series of root-mean-square values of audio signal samples, and then filtering the series of root-mean-square values with a window function to generate a long term root-mean-square average to reduce the effect of sudden amplitude increase in the audio signal.

32. The method of Claim 26, further comprising using the compander to determine a target gain level TG for the gain of the audio signal according to the formula

$$TG = MinG + (GR * \min[1.0, (e - NT) / NR]),$$

where $MinG$ is representative of a minimum gain level, GR is representative of a gain range, e is representative of the error signal level, NT is representative of a noise threshold, and NR is representative of a noise range level.

33. The method of Claim 32, wherein the minimum gain level is approximately between –40 dB and 0 dB.

34. The method of Claim 32, wherein the gain range is approximately between 0 dB and 40 dB.

35. The method of Claim 32, wherein the noise threshold is approximately between –80 dB and 0 dB.

36. The method of Claim 32, wherein the noise range is approximately between 1 dB and 60 dB.

37. The method of Claim 32, further comprising using the compander to adjust the current gain applied to the music signal by increasing the current gain if the target gain is greater than the current gain.

38. The method of Claim 37, further comprising determining the rate of increase in the current gain according to the formula

current gain = current gain + a1 / (Attack Time),

where a1 is a predetermined constant, and Attack Time represents the amount of time for the current gain to increase by 40 dB.

39. The method of Claim 32, further comprising using the compander to adjust the current gain applied to the music signal by decreasing the current gain if the target gain is less than the current gain.

40. The method of Claim 39, wherein the rate of decrease in the current gain is responsive to a user-defined parameter Release Time and determined according to the formula

current gain = current gain – a2 / (Release Time),

where a2 is a predetermined constant, and Release Time represents the amount of time for the current gain to decrease by 40 dB.

41. The method of Claim 26, further comprising storing parameters used by the adaptive filter in a non-volatile memory.

42. The method of Claim 41, further comprising initially determining the adaptive coefficients according to a calibration procedure having an *Override* state that saves a set of parameter values currently used by the compander and in replacement thereof a set of default parameter values optimized for fast adaptation, a *Wait* state that adjusts and monitors the adaptation coefficients until the adaptation coefficients are optimized, a *Compute* state that computes a calibration value for use in preventing gain runaway, a *Save* state that saves the optimized adaptation coefficients in the non-volatile memory device, and a *Restore* state that restores the set of parameters saved during the *Override* state.

43. The method of Claim 26, further comprising the compander to adjust the gain of the audio signal when the error signal is above a predetermined noise threshold level.

44. The method of Claim 43, further comprising defining the predetermined noise threshold level by a user.

45. The method of Claim 44, further comprising defining the user-defined noise threshold to a Noise Threshold Override value when the user-defined noise threshold is lower than the Noise Threshold Override value, wherein the Noise Threshold Override value is computed by:

Noise Threshold Override =
max [(GR / max (CR, 1.0) + MaxG + max (CG, CI) + CAL), NT]

where GR is representative of a gain range, CR is representative of a compression ratio between the gain range and a noise range, MaxG is representative of a maximum gain, CG is representative of a compander gate, CI is representative of a compander input level, CAL is representative of a calibration value, and NT is the user-defined noise threshold.

46. The method of Claim 45, further comprising computing the calibration value *CAL* by adding the RMS value of the error signal and the RMS value of a headroom signal, then subtracting the RMS value of the audio signal.

47. The method of Claim 46, further comprising calculating the RMS value of the headroom signal by taking the difference between the RMS values of the ambient room signal and the error signal, multiplying the difference with a first constant and then subtracting a second constant.

48. The method of Claim 47, wherein the first constant is approximately 0.6, and the second constant is approximately 3.0.